Separation and Analysis of the Musical Signal Noise Components

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Introduction

The noise components of the musical instrument sound may have influence on the timbre of the tone and on the perceived sound quality of the tone. For example, when the tone D6 (f0 = 1175 Hz) is played on the violin, the important noise components falls in the interval 230-800 Hz, that means below the fundamental harmonic [1, 2]. When the low tones are played, the noise components falls in between of the harmonics. The supression of the noise components is sufficient for the perceptual investigation of the noise part influence on the timbre. On the other hand we need extract the noise part from the sound if we want investigate it's physical properties. This process is presented in the example of the violin tone.

Method

At first, the upper and lower bounds of the analysed frequency band are determined. The lower bound is equal to the frequency of the fundamental harmonic. The upper bound is determined by the frequency of the highest harmonic which amplitude falls above the hearing threshold [3], see **Figure 1**.

The methods used for separation are based on the two basic theoretical models of the tone creation. The first one, called **additive model**, assumes the tone as a combination of the set of stable harmonics (sinusoids with constant amplitudes and frequencies) and the noise representing all unstabilities of the signal. This model very well describes the tones played without any pitch changes or *vibrato*.

The second one, called **modulation model**, assumes the tone as the set of sinusoids, which may show pitch changes or frequency modulation resulting from *vibrato*, and the noise. It models the reality more precise. In the very detailed approach the harmonics may be modulated not only by the musician's intentional modulation but also by the modulation function arising from the excitator-oscillator (e.g. bow-string) interaction.

The separation methods consists in the harmonics cancellation. The harmonics must be cancelled as much as possible whereas the noise part must be preserved, that means that the cancelled frequency band must be very narrow. The two problems must have been resolved:

- the accuracy of the harmonic frequency computation is limited mainly by the frequency resolution of the metod used for the spectrum computation. The frequency resolution may be enhanced by the successive application of cubic interpolation [4].
- the frequencies of the harmonics of the naturally played tone fluctuate; the solution of this problem is described in the next part of this paper.

Application of the M-Q Analysis

The first separation method is the implementetion of the modulation model. It was built on the application of the McAuly-Quatieri analysis [5]. This analytical method is applied e.g. in speech processing, musical signal restoration or signal modelling. It is based on the analysis and resynthesis of the signal.

During the analytical phase of this method the time flow of the signal is divided into short time intervals - windows. Then the instantaneous frequency spectrum is computed for each time window. In each spectrum the spectral peaks are detected and they are assumed as a harmonics of the signal. The setup of the detection threshold affect the discrimination of the harmonics and noise. Then the instntaneous parameters (amplitude, frequency and phase shift) are computed for each harmonic.

In the resynthesis phase the parameters are used for generation of the sinusoids from which the resulting signal is additively synthesised. This signal consists of the harmonics without noise.

In our application of the M-Q analysis method the resulting signal is subtracted from the original signal in the time domain (see **Figure 2**). This subtraction acst as an adaptive comb filter with very narrow stop bands and very steep transition bands. This idea is very powerfull, but it's application was not succesfull. The main disadvantage of the application of that method was the limited accuracy of the harmonics parameters computation.

Harmonics Filtration

The implementation of the additive model results in filtration of the harmonics by the non-adaptive comb filter. This method is more simple then the previous one. It force us to resolve two basic problems:

- determination of the harmonic structure of the tone. The frequencies of the harmonics are obtained from the long-term spectrum computed from the quasi-stationary part of the played tone. The accuracy of the computation is enhanced using cubic interpolationin the next step.
- determination of the stopband widths for each harmonic. The first criterion is the frequency fluctuationof the harmonic. The stopband width must be determined so that the harmonics will be filtered in any time instant of the signal time flow. It results in relatively narrow filter stopbands. The filter design algorithm requires to give larger stopband width parameter.

Computation of the filters and signal filtration were done in MATLAB. The direct form type I FIR filters of the order 1800 were used. The first step of the filter design was the determination of the desired filter frequency response, which was derived from the signal harmonic spectrum. Then the constrained least square filter design algorithm was applied. The MATLAB uses the multiple exchange algorithm that uses Lagrange multipliers and Kuhn-Tucker conditions on each iteration step. The algorithm minimizes the square errors among desired and resulting filter frequency response limitating ripple in passbands and stopbands to a given value. Becouse of the limitation of the stopband attenuation the multiple filtration was used (see **Figure 3**).

Discussion

The analyzed signals were the sounds of instruments played by the musician in an anechoic room, which were digtally recorded using the set of high quality studio microphones. The sampling frequency 44100 Hz and 16-bits recording was used. Each recording was calibrated to SPL for further analysis. The acquisited signals were stored in a WAVE files on CD-ROM discs. Processing and analysis is done on PC in the floating point real number format.

The signal spectra were computed using Welch PSD estimation with the window length 4096 samples. In the harmonic spectrum computation the cubic approximation enhances the frequency resolution to value about 0.5 Hz. Then two different methods for the noise part separation were used.

The application of the McAuly-Quatieri analysis was not succesfull. The accuracy of the computation of the parameter values is not sufficient for the signal resynthesis over long time interval. Also the resynthesis algorithm, which computes signal with piecewise-constant parameters over very short time windows is not satisfactory. The future enhancement of this algorithm will adapt parameter values (obtained using interpolation of the parameter values tabeled during the analysis) for each sampling interval. Then we expect smoother resynthesised signal which will better correspond to the harmonic part of the original signal.

This time the filtration method shows better results. The stopbands with the constant frequency width were used in the comb filter design. These filters well covers the small frequency fluctuations of the harmonic in the most cases, but they cannot trap the large frequency changes (modulations or trends). The filters with constant absolute width over the whole frequency band seems to be not ideal mainly for the higher frequencies. The filters with constant relative width results in the very narrow stopbands on the low frequencies or in the wide stopbands on the high frequencies. The further modification will encompass the combination of the constant absolute / constant relative filter width and the method of the individual stopband width for each harmonics.

Acknowledgement

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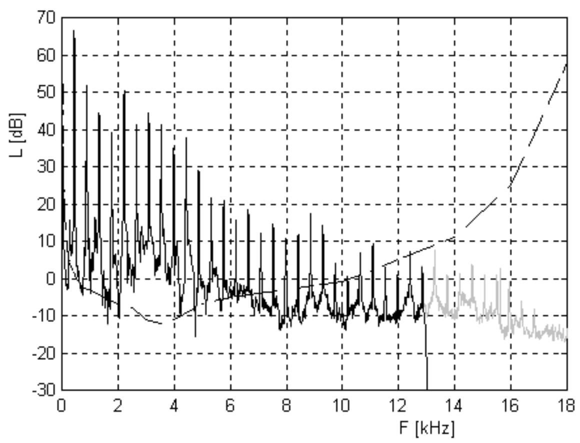

Figure 1: Determination of the upper bound of the analysed frequency band. Gray: spectrum of the signal; black: analysed frequency band; dashed: hearing threshold.

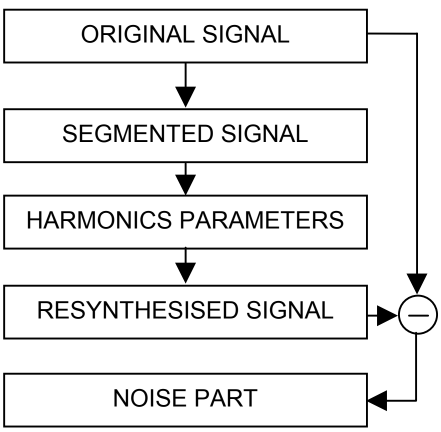

Figure 2: Block scheme of the application of the M-Q analysis for noise part separation

References

[1] ŠTĚPÁNEK, J., OTČENÁŠEK, Z. (1999): Rustle as an Attribute of Timbre of Stationary Violin Tones, CASJ - Journal of the CATGUT Acoustical Society, Vol. 3, No. 8 (Series II), November 1999, 32-38.

[2] ŠTĚPÁNEK, J., OTČENÁŠEK, Z., MORAVEC, O. (2000): Analytical and Perceptual Detection of Rustle in Stationary Violin Tones, SFA / 5 CFA, Lausanne 2000, 433-436.

[3] ZWICKER, E., FELDTKELLER, R.: The Ear as a Communication Receiver. American Institute of Physics, Translation for JASA, 1999; p. 68

[4] - MATLAB (1996): Signal Processing Toolbox User's Guide, The MathWorks, Inc.

[5] MCAULY, J. R., QUATIERI, T. F. (1986): Speech Analysis/Synthesis Based on a Sinusoidal Representation. *IEEE Transactions on Acoustics, Speech and Signal Processing*, vol. ASSP-34, 744 – 754.

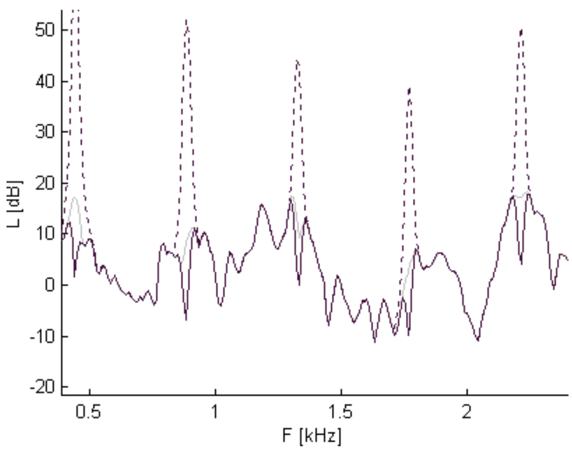

Figure 3: Effect of the double filtration of the harmonics. dotted: original; gray: 1-st filtration; black: 2-nd filtration